# High Performance Networking

SC2002 Tutorial M12 18 November 2002

Phillip Dykstra
Chief Scientist
WareOnEarth Communications Inc.
phil@sd.wareonearth.com

### Motivation

If our networks are so fast, how come my ftp is so slow?

Dykstra, SC2002

### **Objectives**

- Look at current high performance networks
- Learn what is required for high speed data transfer and what to expect
- Fundamental understanding of delay, loss, bandwidth, routes, MTU, windows
- Look at useful tools and what they tell you
- Examine TCP dynamics and TCP's future

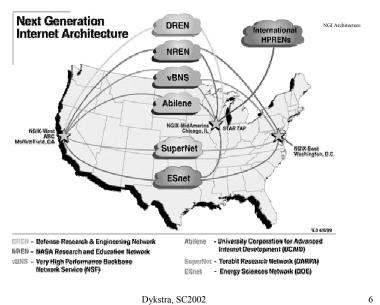
Dykstra, SC2002

### Unique HPC Environment

- The Internet is being optimized for:
  - millions of users behind low-speed soda straws
  - thousands of high-bandwidth servers serving millions of soda straw streams
- Single high-speed to high-speed flows get little commercial attention

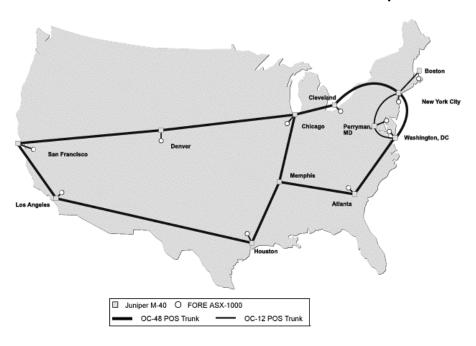
Dykstra, SC2002

# High Performance Networks in the USA

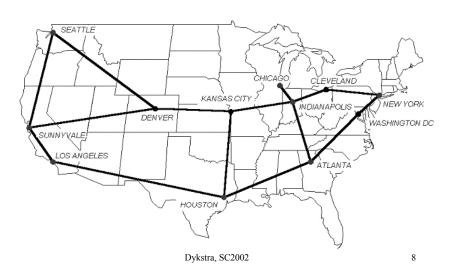


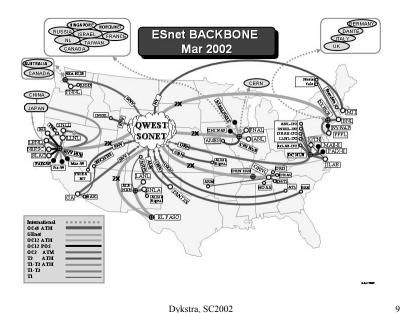
Dykstra, SC2002

### vBNS+ Backbone Network Map

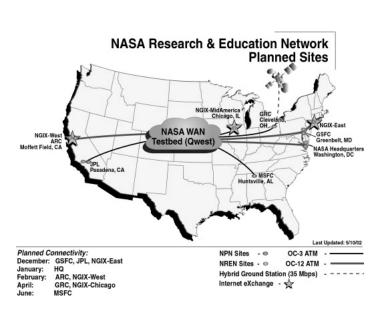


#### Abilene Network Backbone - February 2002



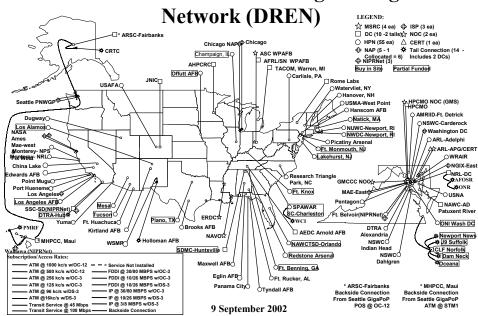


Dykstra, SC2002

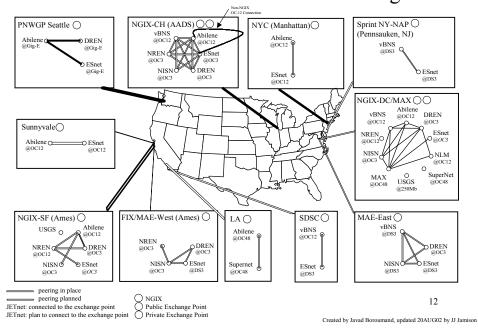


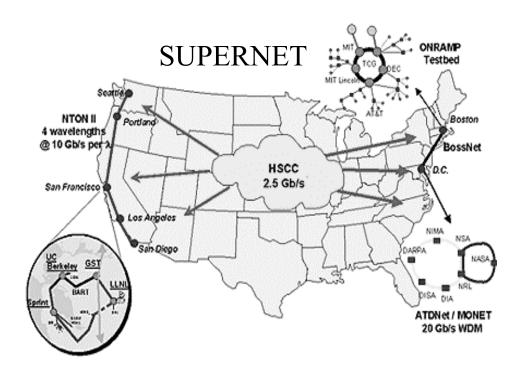
Dykstra, SC2002

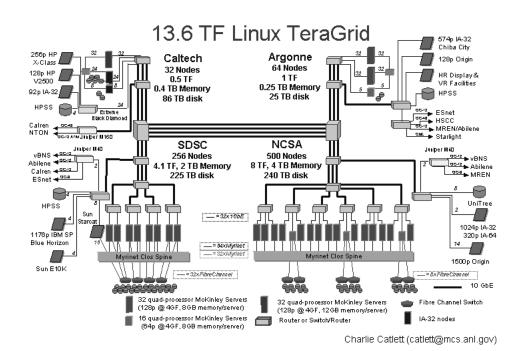
#### **Defense Research and Engineering**



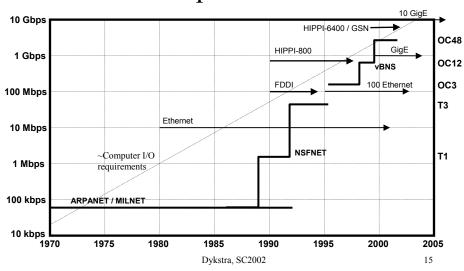
#### JETnets Interconnections and Peering







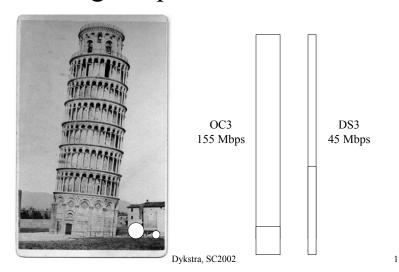
# Network Speeds Over Time



## Delay

a.k.a. Latency

# Capacity High "Speed" Networks



### Speed of Light in Media

- $\sim 3.0 \times 10^8$  m/s in free space
- $\sim 2.3 \times 10^8$  m/s in copper
- $\sim 2.0 \times 10^8$  m/s in fiber = 200 km / ms [100 km of distance = 1 ms of round trip time]

Dykstra, SC2002

### Packet Durations and Lengths

1500 Byte Packets in Fiber

	Mbps	pps	sec/pkt	lengt	h
56k	0.056	4.7	214 ms	42857	km
T1	1.544	129	7.8 ms	1554	km
Eth	10	833	1.2 ms	240	km
Т3	45	3750	267 us	53	km
FEth	100	8333	120 us	24	km
oc3	155	13k	77 us	: 15	km
OC12	622	52k	19 us	3859	m
GigE	1000	83k	12 us	2400	m
OC48	2488	207k	4.8 us	965	m
10GigE	10000	833k	1.2 us	240	m

Dykstra, SC2002

19

## Observations on Packet Lengths

• A 56k packet could wrap around the earth!



• A 10GigE packet fits in the convention center

Dykstra, SC2002

### Observations on Packet Lengths

- Each store and forward hop adds the packet duration to the delay
  - In the old days (< 10 Mbps) such hops dominated delay
  - Today (> 10 Mbps) store and forward delays on WANs are minimal compared to propagation

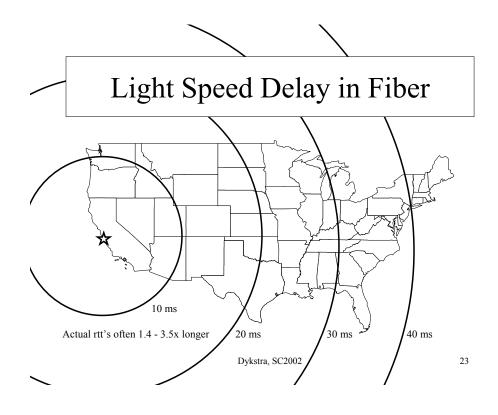
Dykstra, SC2002

21

### Observations on Packet Lengths

- ATM cells (and TCP ACK packets) are  $\sim 1/30^{th}$  as long, 30x as many per second
  - One of the reasons we haven't seen OC48 SAR until now (2002)
- Jumbo Frames (9000 bytes) are 6x longer, 1/6<sup>th</sup> as many per second

Dykstra, SC2002



### Measuring Delay - Ping

```
% ping -s 56 sgi.com
PING sgi.com (192.48.153.65) from 63.196.71.246 : 56(84) bytes of data.
64 bytes from SGI.COM (192.48.153.65): icmp_seq=1 ttl=240 time=31.6 ms
64 bytes from SGI.COM (192.48.153.65): icmp_seq=2 ttl=240 time=66.9 ms
64 bytes from SGI.COM (192.48.153.65): icmp_seq=3 ttl=240 time=33.4 ms
64 bytes from SGI.COM (192.48.153.65): icmp_seq=4 ttl=240 time=36.7 ms
64 bytes from SGI.COM (192.48.153.65): icmp_seq=5 ttl=240 time=40.9 ms
64 bytes from SGI.COM (192.48.153.65): icmp_seq=6 ttl=240 time=104.8 ms
64 bytes from SGI.COM (192.48.153.65): icmp_seq=6 ttl=240 time=177.5 ms
64 bytes from SGI.COM (192.48.153.65): icmp_seq=8 ttl=240 time=34.2 ms
64 bytes from SGI.COM (192.48.153.65): icmp_seq=8 ttl=240 time=31.5 ms
64 bytes from SGI.COM (192.48.153.65): icmp_seq=9 ttl=240 time=31.5 ms
64 bytes from SGI.COM (192.48.153.65): icmp_seq=10 ttl=240 time=31.9 ms
--- sgi.com ping statistics ---
11 packets transmitted, 10 packets received, 9% packet loss
round-trip min/avg/max = 31.5/58.9/177.5 ms
```

Dykstra, SC2002

### Ping Observations

IP 20	8	User Data 0+ bytes
	ICM	p

- Ping packet = 20 bytes IP + 8 bytes ICMP
  + "user data" (first 8 bytes = timestamp)
- Default = 56 user bytes = 64 byte IP payload = 84 total bytes
- Small pings (-s 8 = 36 bytes) take less time than large pings (-s 1472 = 1500 bytes)

Dykstra, SC2002

### **Ping Observations**

- TTL = 240 indicates 255-240 = 15 hops
- Delay variation indicates congestion or system load
- Not good at measuring small loss
  - An HPC network should show zero ping loss
- Depends on ICMP ECHO which is sometimes blocked for "security"

Dykstra, SC2002 26

### Bandwidth\*Delay Product

- The number of bytes in flight to fill the entire path
- Includes data in queues if they contributed to the delay
- Example
  - 100 Mbps path
  - ping shows a 75 ms rtt
  - -BDP = 100 \* 0.075 = 7.5 million bits (916 KB)

Dykstra, SC2002

27

#### Routes

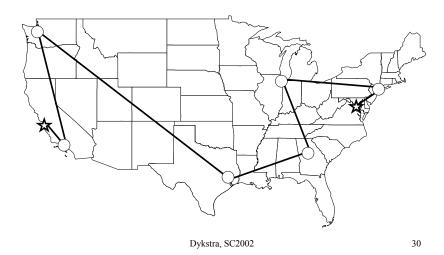
The path taken by your packets

#### **How Routers Choose Routes**

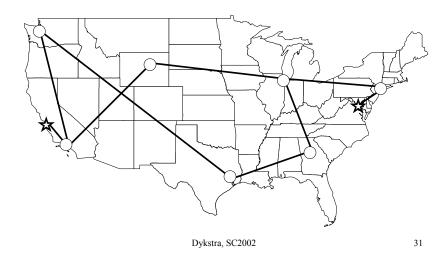
- Within a network
  - Smallest number of hops
  - Highest bandwidth paths
  - Usually ignore latency and utilization
- From one network to another
  - Often "hot potato" routing, i.e. pass to the other network ASAP

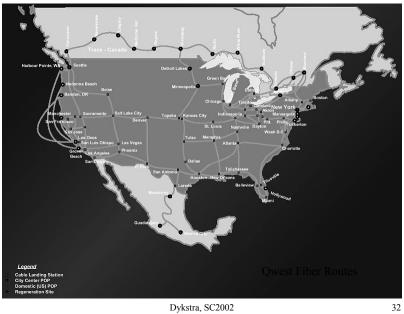
Dykstra, SC2002

### "Scenic" Routes

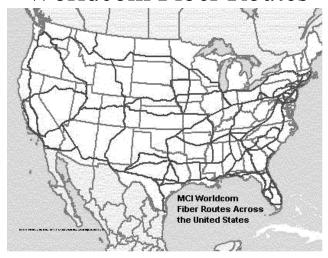


# **Asymmetric Routes**



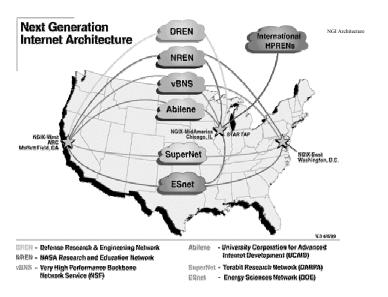


### Worldcom Fiber Routes



Dykstra, SC2002

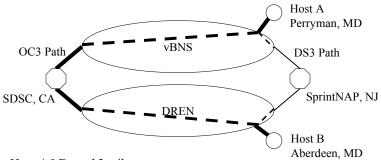
3:



Dykstra, SC2002

#### Path Performance: Latency vs. Bandwidth

The highest bandwidth path is not always the highest throughput path!



- Host A&B are 15 miles apart
- DS3 path is ~250 miles
- OC3 path is ~6000 miles

The network chose the OC3 path with 24x the rtt, 80x BDP

Dykstra, SC2002

35

#### How Traceroute Works

www.caida.org/outreach/resources/animations/

- Sends UDP packets to ports (-p) 33434 and up, TTL of 1 to 30
- Each router hop decrements the TTL
- If the TTL=0, that node returns an ICMP TTL Expired
- The destination host returns an ICMP Port Unreachable

Dykstra, SC2002

#### **Traceroute Observations**

- Shows the return interface addresses of the **forwarding** path
- You can't see hops through switches or over tunnels (e.g. ATM VC's, GRE, MPLS)
- The required ICMP replies are sometimes blocked for "security", or not generated, or sent without resetting the TTL

Dykstra, SC2002 37

#### Matt's Traceroute

#### www.bitwizard.nl/mtr/

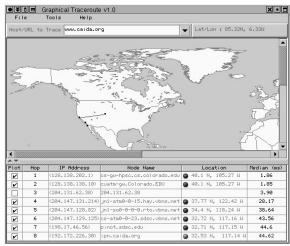
Matt's traceroute [v0.41]												
damp-ssc.spawar.navy.mil	Sun Apr 23 23:29:51 2000											
Keys: D - Display mode R - Restart	statistics Q - Quit											
	Packets			Pings								
Hostname	%Loss	Rcv	Snt	Last	Best	Avg	Worst					
1. taco2-fe0.nci.net	0%	24	24	0	0	0	1					
<ol><li>nccosc-bgp.att-disc.net</li></ol>	0%	24	24	1	1	1	6					
<ol><li>pennsbr-aip.att-disc.net</li></ol>	0%	24	24	84	84	84	86					
<ol> <li>sprint-nap.vbns.net</li> </ol>	0%	24	24	84	84	84	86					
<ol><li>cs-hssi1-0.pym.vbns.net</li></ol>	0%	23	24	89	88	152	407					
6. jn1-at1-0-0-0.pym.vbns.net	0%	23	23	88	88	88	90					
7. jn1-at1-0-0-13.nor.vbns.net	0%	23	23	88	88	88	90					
8. jn1-so5-0-0-0.dng.vbns.net	0%	23	23	89	88	91	116					
9. jn1-so5-0-0-0.dnj.vbns.net	0%	23	23	112	111	112	113					
10. jn1-so4-0-0-0.hay.vbns.net	0%	23	23	135	134	135	135					
11. jn1-so0-0-0-0.rto.vbns.net	0%	23	23	147	147	147	147					
12. 192.12.207.22	5%	22	23	98	98	113	291					
13. pinot.sdsc.edu	0%	23	23	152	152	152	156					
14. ipn.caida.org	0%	23	23	152	152	152	160					

Dykstra, SC2002

38

## GTrace – Graphical Traceroute

www.caida.org/tools/visualization/gtrace/



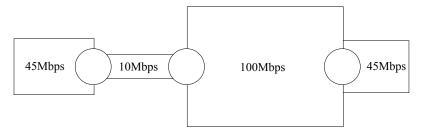
Dykstra, SC2002

39

### Bandwidth

and throughput

### Hops of Different Bandwidth



- The "Narrow Link" has the lowest bandwidth
- The "Tight Link" has the least Available bandwidth
- Queues can form wherever available bandwidth decreases
- A queue buildup is most likely in front of the Tight Link

Dykstra, SC2002

41

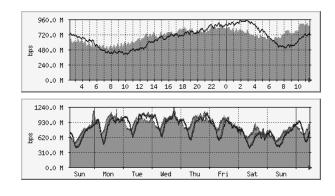


- www.mrtg.org
- Extremely popular network monitoring tool
- Most common display:
  - Five minute average link utilizations
  - Green into interface
  - Blue out of interface
- RRDTool newer generalized version (same site)

Dykstra, SC2002

# MRTG Example

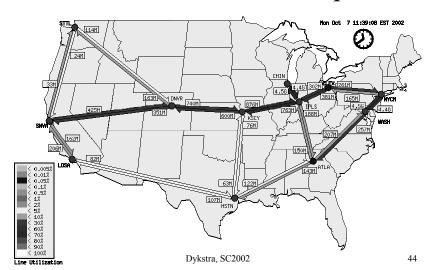
Abilene, Indianapolis to Kansas City, OC48 link, 7 October 2002



Dykstra, SC2002

43

# Abilene "Weather Map"



### Throughput Limit

• throughput <= available bandwidth

("tight link" with the minimum unused bandwidth)

- A high performance network should be lightly loaded (<50%)</li>
- A loaded high speed network is no better to the end user than a lightly loaded slow one

Dykstra, SC2002 4

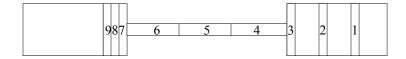
#### Bandwidth Estimation – Single Packet

#### 

- Larger packets take longer
- · Delay from intercept
- · Bandwidth from slope

From A. Downey

#### Bandwidth Estimation – Multi Packet



- · Packet pairs or trains are sent
- The slower link causes packets to spread
- The packet spread indicates the bandwidth

Dykstra, SC2002

#### **Bandwidth Measurement Tools**

- pathchar Van Jacobson, LBL
  - <u>ftp://ftp.ee.lbl.gov/pathchar/</u>
- clink Allen Downey, Wellesley College
  - $\, \underline{http://rocky.wellesley.edu/downey/clink/}$
- pchar Bruce A. Mah, Sandia/Cisco
  - $\underline{http://www.employees.org/\!\!\sim\!\!bmah/Software/pchar/}$

Dykstra, SC2002

#### **Bandwidth Measurement Tools**

- pipechar Jin Guojun, LBL
  - http://www.didc.lbl.gov/pipechar/
- nettimer Kevin Lai, Stanford University
  - <u>http://gunpowder.stanford.edu/~laik/projects/nettimer/</u>
- pathrate/pathload Constantinos Dovrolis, Georgia Tech
  - http://www.cc.gatech.edu/fac/Constantinos.Dovrolis/bwmeter.html

Dykstra, SC2002

49

#### Treno Throughput Test

www.psc.edu/networking/treno\_info.html

 Tells you what a good TCP should be able to achieve (Bulk Transfer Capacity)

#### damp-mhpcc% treno damp-pmrf

MTU=8166 MTU=4352 MTU=2002 MTU=1492 ......

Replies were from damp-pmrf [192.168.1.1]

Average rate: 63470.5 kbp/s (55241 pkts in + 87 lost = 0.16%) in 10.03 s Equilibrium rate: 63851.9 kbp/s (54475 pkts in + 86 lost = 0.16%) in 9.828 s

Path properties: min RTT was 8.77 ms, path MTU was 1440 bytes

Dykstra, SC2002 50

#### Treno Observations

- Easy 10 second test, no remote access or receiver process required
- Emulates TCP but doesn't use TCP
  - Problems with host TCP or tuning are avoided
- Does Path MTU Discovery
- Reports rtt and loss rates
- A zero equilibrium result means there was too much packet loss to exit "slow start"

Dykstra, SC2002

5

#### Treno Observations

- Can send ICMP (-i) or UDP (default)
  - ICMP replies (ECHO or UNREACH) could be blocked for "security"
- Routers send ICMP replies very slowly
  - So don't test routers with treno
- ICMP is often rate limited now by hosts
- Port numbers can wrap around (and look like port scans)

Dykstra, SC2002

### TCP Throughput Tests

- ttcp the original, many variations
  - http://sd.wareonearth.com/~phil/net/ttcp/
- nuttcp great successor to ttcp (recommended)
  - ftp://ftp.lcp.nrl.navy.mil/u/bill/beta/nuttcp/
- Iperf great TCP/UDP tool (recommended)
  - <a href="http://dast.nlanr.net/Projects/Iperf/">http://dast.nlanr.net/Projects/Iperf/</a>
- netperf dated but still in wide use
  - http://www.netperf.org/
- ftp nothing beats a real application

Dykstra, SC2002

5.

#### nuttep: TCP, 10ms rtt, OC12 path

```
damp-arl$ nuttcp -r -T10 -w1000 damp-asc2-atm
nuttcp-r: v3.1.9: socket
nuttcp-r: buflen=65536, nstream=1, port=5001 tcp
nuttcp-r: accept from 192.168.131.9
nuttcp-r: send window size = 65536, receive window size = 2048000
nuttcp-r: 604.860 MB in 10.01 real seconds = 61895.32 KB/sec = 507.0464 Mb/s
nuttcp-r: 69593 I/O calls, msec/call = 0.15, calls/sec = 6954.54
nuttcp-r: 0.0user 0.7sys 0:10real 8% 0i+0d 0maxrss 0+0pf 0+0csw
nuttcp-t: v3.1.9: socket
nuttcp-t: buflen=65536, nstream=1, port=5001 tcp -> 140.32.131.13
nuttcp-t: time limit = 10 seconds
nuttcp-t: connect to 192.168.131.13
nuttcp-t: send window size = 2048000, receive window size = 349520
nuttcp-t: 604.860 MB in 10.00 real seconds = 61958.83 KB/sec = 507.5667 Mb/s
nuttcp-t: 9678 I/O calls, msec/call = 1.06, calls/sec = 968.13
nuttcp-t: 0.0user 3.3sys 0:10real 33% 0i+0d 0maxrss 0+0pf 0+0csw
                                Dykstra, SC2002
                                                                        54
```

### Throughput Testing Notes

- Network data rates (bps) are powers of 10, not powers of 2 as used for Bytes
  - E.g. 100 Mbps ethernet is 100,000,000 bits/sec
  - Some tools wrongly use powers of 2 (e.g. ttcp)
- User payload data rates are reported by tools
  - No TCP, IP, Ethernet, etc. headers are included
  - E.g. 100 Mbps ethernet max is 97.5293 Mbps
    - <a href="http://sd.wareonearth.com/~phil/net/overhead/">http://sd.wareonearth.com/~phil/net/overhead/</a>

Dykstra, SC2002

55

#### Windows

Flow/rate control and error recovery

#### Windows

- Windows control the amount of data that is allowed to be "in flight" in the network
- Maximum throughput is one window full per round trip time
- The sender, receiver, and the network each determine a different window size

Dykstra, SC2002

57

### Window Sizes 1,2,3

Data packets go one way ACK packets come back

Dykstra, SC2002

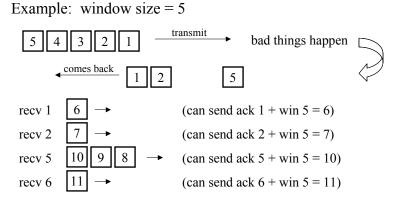
### MPing – A Windowed Ping

www.psc.edu/~mathis/wping/

- Excellent tool to view the packet forwarding and loss properties of a path under varying load!
- Sends windows full of ICMP Echo or UDP packets
- Treats ICMP Echo\_Reply or Port\_Unreachable packets as "ACKs"
- Make sure destination responds well to ICMP
- Consumes a lot of resources: use with care

Dykstra, SC2002

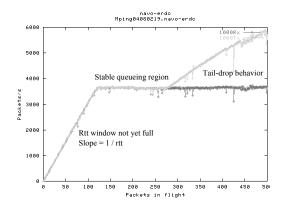
### How MPing Works



Dykstra, SC2002

60

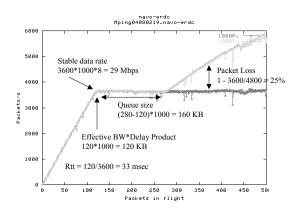
# MPing on a "Normal" Path



Dykstra, SC2002

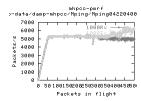
61

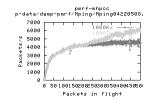
# MPing on a "Normal" Path



Dykstra, SC2002

### Some MPing Results #1



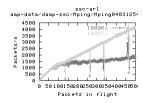


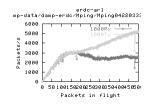
Fairly normal behavior Discarded packets are costing some performance loss RTT is increasing as load increases Slow packet processing?

Dykstra, SC2002

63

### Some MPing Results #2

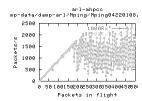




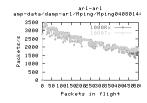
Very little stable queueing Insufficient memory? Spikes from some periodic event (cache cleaner?) Discarding packets comes at some cost to performance Error logging?

Dykstra, SC2002

### Some MPing Results #3



Oscillations with little loss Rate shaping?

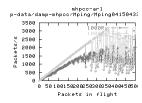


Decreasing performance with increasing queue length Typical of Unix boxes with poor queue insertion

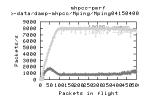
Dykstra, SC2002

65

## Some MPing Results #4



Fairly constant packet loss, even under light load



Major packet loss, ~7/8 or 88% Hump at 50 may be duplex problem

Both turned out to be an auto-negotiation duplex problem Setting to static full-duplex fixed these!

Dykstra, SC2002

### TCP Throughput

(window/rtt)

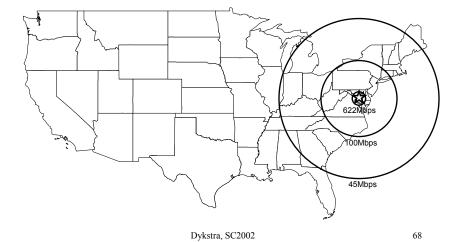
- The smallest of three windows determines throughput
  - sbuf, or sender side socket buffers
  - rwin, the receive window size
  - cwin, TCP congestion window
- Receive window (rwin) and/or sbuf are still the most common performance limiters
  - E.g. 8kB window, 87 msec ping time = 753 kbps
  - E.g. 64kB window, 14 msec rtt = 37 Mbps

Dykstra, SC2002

67

### Maximum TCP/IP Data Rate

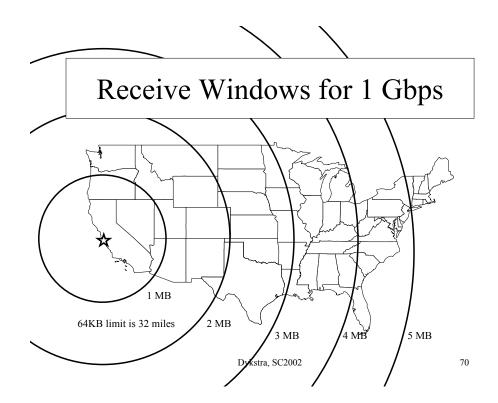
With 64KB window

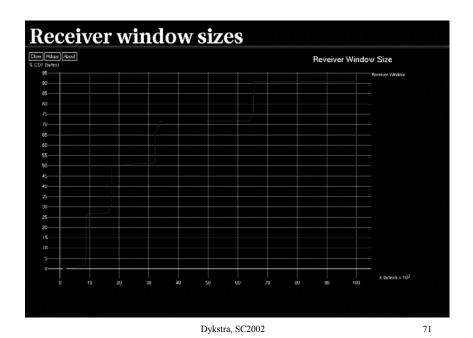


#### Bandwidth\*Delay Product and TCP

- TCP needs a **receive window** (rwin) equal to or greater than the BW\*Delay product to achieve maximum throughput
- TCP needs **sender side socket buffers** of 2\*BW\*Delay to recover from errors
- You need to send about 3\*BW\*Delay bytes for TCP to reach maximum speed

Dykstra, SC2002





# Observed Receiver Window Sizes

- ATM traffic from the Pittsburgh Gigapop
- 50% have windows < 20 KB
  - These are obsolete systems!
- 20% have 64 KB windows
  - Limited to  $\sim 8$  Mbps coast-to-coast
- ~9% are assumed to be using window scale

M. Mathis, PSC

Dykstra, SC2002

## System Tuning

Interfaces, routes, buffers, etc.

## Things You Can Do



- Throw out your low speed interfaces and networks!
- Make sure routes and DNS report high speed interfaces
- Don't over-utilize your links (<50%)
- Use routers sparingly, host routers not at all routed -q

Dykstra, SC2002

## Things You Can Do



- Make sure your HPC apps offer sufficient receive windows and use sufficient send buffers
  - But don't run your system out of memory
  - Find out the rtt with ping, compute BDP
  - Can tune system wide, by application, or automatically
- Check your TCP for high performance features
- Look for sources of loss
  - Watch out for duplex problems (late collisions?)

Dykstra, SC2002

7:

## System Tuning: Linux 2.4

#### /etc/sysctl.conf

```
# Increase max socketbuffer sizes, actual = 2x these values
net.core.rmem max = 1048576
net.core.wmem max = 1048576
# Increase min, default, auto buffer sizes
net.ipv4.tcp rmem = 4096
                             349520 699040
net.ipv4.tcp wmem = 4096
                              65536 524288
# Remove ICMP reply rate limits (now icmp_ratemask)
net.ipv4.icmp echoreply rate = 0
net.ipv4.icmp_destunreach_rate = 0
# Misc performance features (defaults)
net.ipv4.ip\_no\_pmtu\_disc = 0
net.ipv4.tcp\_sack = 1
net.ipv4.tcp\_window\_scaling = 1
net.ipv4.tcp\_timestamps = 1
net.ipv4.tcp ecn = 0
```

Dykstra, SC2002

#### System Tuning: FreeBSD

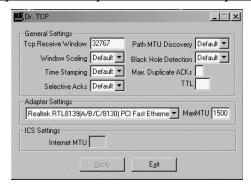
# FreeBSD 3.4 defaults are 524288 max, 16384 default /sbin/sysctl -w kern.ipc.maxsockbuf=1048576 /sbin/sysctl -w net.inet.tcp.sendspace=32768 /sbin/sysctl -w net.inet.tcp.recvspace=32768

Dykstra, SC2002

77

#### Dr. TCP

A TCP Stack Tuner for Windows <a href="http://www.dslreports.com/front/drtcp.html">http://www.dslreports.com/front/drtcp.html</a>



- See also, <a href="http://cable-dsl.home.att.net/">http://cable-dsl.home.att.net/</a> for registry edits, etc.
- Beware that modem utilities such as DunTweak can reduce performance on high speed nets

Dykstra, SC2002

## **Ethernet Duplex Problems**

#### An Internet Epidemic!

- Ethernet "auto-negotiation" can select the speed and duplex of a connected pair
- If only one end is doing it:
  - It can get the speed right
  - but it will assume half-duplex
- Mismatch loss only shows up under load
  - Can't see it with ping

Dykstra, SC2002

79

## **Tuning FTP**

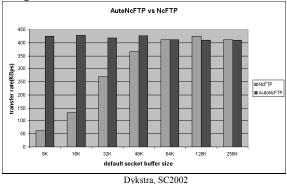
- Many FTP's allow the user to set buffer sizes
- The commands are different everywhere!
- See <a href="http://dast.nlanr.net/Projects/FTP.html">http://dast.nlanr.net/Projects/FTP.html</a> for several sets of commands

Dykstra, SC2002

## Autobuf – An Auto-tuning FTP

http://dast.nlanr.net/Projects/Autobuf/

 Measures the spread of a burst of ICMP Echo packets to estimate BDP, sets bufs



8

## Good Tuning References

- Users Guide to TCP Windows
   www.ncsa.uiuc.edu/People/vwelch/net\_perf/tcp\_
   windows.html
- TCP Tuning Guide www-didc.lbl.gov/TCP-tuning/
- Enabling High Performance Data Transfers on Hosts

www.psc.edu/networking/perf tune.html

Dykstra, SC2002

## TCP

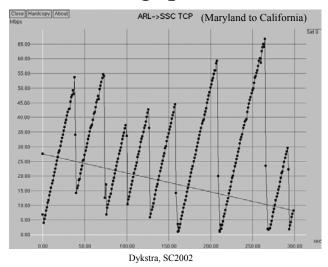
The Internet's transport

## Important Points About TCP

- TCP is adaptive
- It is *constantly* trying to go *faster*
- It slows down when it detects a loss
- How much it sends is controlled by windows
- When it sends is controlled by received ACK's (or timeouts)

Dykstra, SC2002

## TCP Throughput vs. Time



85

#### **AIMD**

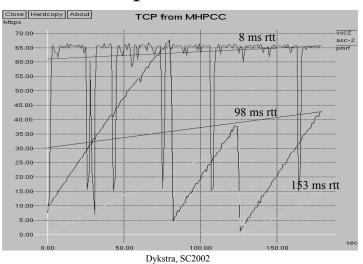
- Additive Increase, Multiplicative Decrease
  - of cwin, the congestion window
- The core of TCP's congestion avoidance phase, or "steady state"
  - Standard increase = +1.0 MSS per loss free rtt
  - Standard decrease = \*0.5 (i.e. halve cwin on loss)
- Avoids congestion collapse
- Promotes fairness among flows

Dykstra, SC2002

## Iperf: TCP California to Ohio

## TCP Examples from Maui HI

Dykstra, SC2002



## TCP Acceleration (MSS/rtt²)

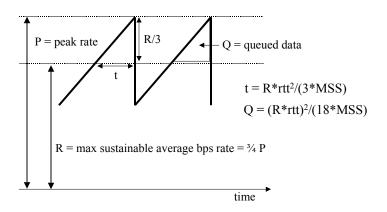
(Congestion avoidance rate increase, MSS = 1448)

rtt (msec)	Mbps/s	0-100Mbps (sec)
5	463	0.216
10	116	0.864
20	29	3.45
50	4.6	21.6
100	1.16	86.4
200	0.29	345

Dykstra, SC2002

89

## TCP Average Rate



Dykstra, SC2002

#### Observations

- Low latency really helps performance!
- Queued data is proportional to the square of the bandwidth delay product (BDP<sup>2</sup>)
  - 100 KB BDP needs ~400 KB queue
  - 1 MB BDP needs ~40 MB queue!
- Queue duration is proportional to the square of the round trip time (rtt²)
- Jumbo frames help **reduce** the size and duration of queues by increasing the slope (TCP acceleration)

Dykstra, SC2002

91

#### Observations

- For high BDP's, average rate will be at best three fourths of the tight link rate
- To do better requires throttling the sender in a way that doesn't halve *cwin* at peaks, e.g.
  - carefully setting *rwin*
  - pacing the sender
  - TCPW (sets cwin based on estimated BW)

#### TCP Throughput

#### www.psc.edu/networking/papers/model\_abstract.html

Once recv window size and available bandwidth aren't the limit

Rate = 
$$\frac{\sim 0.7 * \text{Max Segment Size (MSS)}}{\text{Round Trip Time (latency)} \quad \text{sqrt[pkt\_loss]}_{\text{M. Mathis, et al.}}}$$

- Double the MTU, double the throughput
- Halve the latency, double the throughput
  - shortest path matters
- Halve the loss rate, 40% higher throughput

Dykstra, SC2002

93

### Max Segment Size (MSS)

rate = 0.7 \* MSS / (rtt \* sqrt(p))

- MSS = MTU packet headers
- Common MTU's

576 IPv4 default

1500 ethernet, IPv6 default

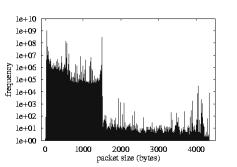
~9000 GigE Jumbo Frame, CLIP ATM

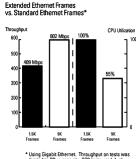
64k max ATM AAL5 frame

• Jumbo frame  $\Rightarrow$   $\sim$ 6x throughput increase

Dykstra, SC2002

#### Packet Size (MTU) Issues





#### http://sd.wareonearth.com/~phil/jumbo.html

"New York to Los Angeles. Round Trip Time (rtt) is about 40 msec, and let's say packet loss is 0.1% (0.001). With an MSS of 1460 bytes, TCP throughput will have an upper bound of about 6.5 Mbps! And no, that is not a window size limitation, but rather one based on TCP's ability to detect and recover from congestion (loss). With 9000 byte frames, TCP throughput could reach about 40 Mbps."

Dykstra, SC2002

95

#### Path MTU

- Maximum Transmission Unit (MTU)
  - Largest packet that can be sent as a unit
- Path MTU (PMTU)
  - min MTU of all hops in a path
- Hosts can do Path MTU Discovery to find it
  - Depends on ICMP replies
- Without PMTU Discovery hosts should assume PMTU is only 576 bytes
  - Some hosts falsely assume 1500!

Dykstra, SC2002

## Things You Can Do



- Use only large MTU interfaces/routers/links
  - Gigabit Ethernet with **Jumbo Frames** (9000)
  - ATM CLIP (9180)
  - Packet over SONET (POS) (4470, 9000+)
- Never reduce the MTU (or bandwidth) on the path between each/every host and the WAN
- Make sure your TCP uses Path MTU Discovery

Dykstra, SC2002

97

#### Round Trip Time (RTT)

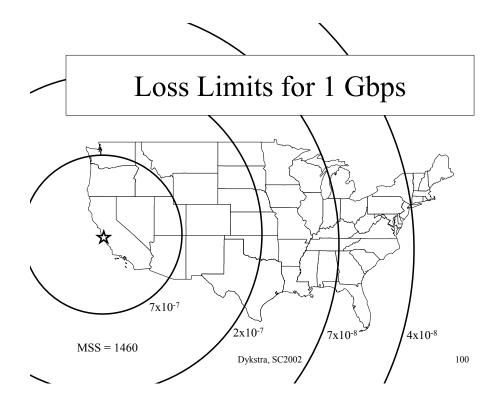
rate = 0.7 \* MSS / (rtt \* sqrt(p))

- If we could halve the delay we could double throughput!
- Most delay is caused by speed of light in fiber (~200 km/msec)
- "Scenic routing" and fiber paths raise the minimum
- Congestion (queueing) adds delay

Dykstra, SC2002

## Packet Loss (p) rate = 0.7 \* MSS / (rtt \* sqrt(**p**))

- Loss dominates throughput!
- At least 6 orders of magnitude observed on the Internet
- 100 Mbps throughput requires O(10<sup>-6</sup>)
- 1 Gbps throughput requires O(10<sup>-8</sup>)



#### Specifying Loss

- TCP loss limits for 1 Gbps across country are O(10<sup>-8</sup>), i.e. 0.000001% packet loss
  - About 1 "ping" packet every three years
  - Systems like AMP would never show loss
  - Try to get 10-8 in writing from a provider!
  - Most providers won't guarantee < 0.01%

Dykstra, SC2002

101

## Specifying Throughput

- Require the provider to demonstrate TCP throughput
  - New DREN RFP requires ½ line rate TCP flow sustained for 10 minutes cross country (e.g. ~300 Mbps on OC12)
- A low loss requirement comes with this!

Dykstra, SC2002

#### Concerns About Bit Errors

- Bit Error Rate (BER) specs for networking interfaces/circuits may not be low enough
  - $E.g. 10^{-12} BER => 10^{-8} packet ER (1500 bytes)$
  - $-10 \text{ hops} => 10^{-7} \text{ packet drop rate}$
- CRC32 and checksums may be too weak
- Error detection in the NIC may miss some
  - E.g. DMA errors (Partridge et al.)

Dykstra, SC2002

103

#### More About TCP

Some details

#### TCP Keeps Evolving

- TCP, RFC 793, Sep 1981
- Reno, BSD, 1990
- Path MTU Discovery, RFC 1191, Nov 1990
- Window Scale, PAWS, RFC 1323, May 1992
- SACK, RFC 2018, Oct 1996
- NewReno, RFC 2581, April 1999
- D-SACK, RFC 2883, July 2000
- More on the way!

Dykstra, SC2002

105

#### TCP Reno

- Most modern TCP's are "Reno" based
- Reno defined (refined) four key mechanisms
  - Slow Start
  - Congestion Avoidance
  - Fast Retransmit
  - Fast Recovery
- NewReno refined fast retransmit/recovery when partial acknowledgements are available

Dykstra, SC2002

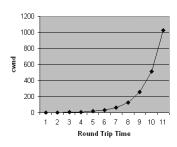
## TCP Congestion Window (cwin)

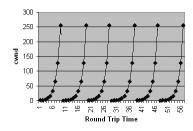
- Congestion window controls startup and limits throughput in the face of loss
- cwin gets larger after every new ACK
- · cwin get smaller when loss is detected
- cwin amounts to TCP's estimate of the available bandwidth at any given time

Dykstra, SC2002

107

## **Cwin During Slowstart**

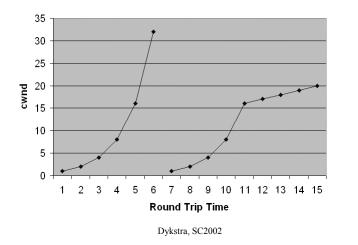




- cwin increased by one for every new ACK
- cwin doubles every round trip time (exponential)
- cwin is reset to zero after a loss

Dykstra, SC2002

# Slowstart and Congestion Avoidance Together

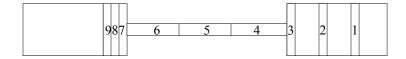


Delayed ACKs

- TCP receivers send ACK's:
  - after every second segment
  - after a delayed ACK timeout
  - on every segment after a loss (missing segment)
- A new segment sets the delayed ACK timer
  - Typically 0-200 msec
- A second segment (or timeout) triggers an ACK and clears the delayed ACK timer

Dykstra, SC2002

## **ACK Clocking**



- A queue forms in front of a slower speed link
- The slower link causes packets to spread
- The spread packets result in spread ACK's
- The spread ACK's end up clocking the source packets at the slower link rate

Dykstra, SC2002 111

#### **Detecting Loss**

- Packets get discarded when queues are full (or nearly full)
- Duplicate ACK's get sent after missing or out of order packets
- Most TCP's retransmit after the third duplicate ACK ("triple duplicate ACK")
  - Windows XP now uses 2<sup>nd</sup> dup ACK

Dykstra, SC2002

#### Random Early Detection (RED)

RFC 2309, Apr 1998

- Discards arriving packets as a function of queue length
- Gives TCP better congestion indications (drops)



- Avoids "Global Synchronization"
- Increases total number of drops
- Increases link utilization
- Many variations (weighted, classed, etc.)

Dykstra, SC2002

113

#### SACK TCP

#### Selective Acknowledgement

- SACK specifies exactly which bytes were missed
- D-SACK can specify which bytes were duplicated
- Better measures the "right edge" of the congestion window (i.e. most recently received data)
- Can do a very good job keeping your queues full
  - Which causes latencies to go way up
- Without RED, will cause global sync faster
- Win98, Win2k, Linux have SACK

Dykstra, SC2002

## Things You Can Do



- Consider using RED on your routers before wide scale deployment of SACK TCP
- SACK won't care very much but your old TCP's will thank you
- Consider a priority class of service for interactive traffic?

Dykstra, SC2002

115

## Advanced Debugging

TCP Traces and Testrig

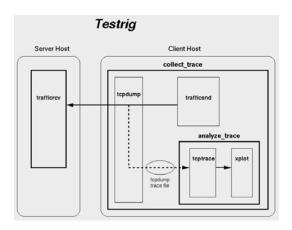
## TCP/IP Analysis Tools

- tcpdump
  - www.tcpdump.org
- ethereal GUI tcpdump (protocol analyzer)
  - www.ethereal.com
- tcptrace stats/graphs of tcpdump data
  - www.tcptrace.org
- testrig tcpdump, tcptrace, xplot, etc.
  - www.ncne.nlanr.net/research/tcp/testrig/

Dykstra, SC2002

117

## "A Preconfigured TCP Test Rig"



Dykstra, SC2002

host b: complete conn: first packet: last packet: elapsed time: total packets:	Sun Apr 23 23:35:29.645263 2000 Sun Apr 23 23:35:41.108465 2000 0:00:11.463202			tcptrace -1		
a->b:		b->a:				
total packets:	72032		total packets:	35793		
ack pkts sent:	72031		ack pkts sent:	35793		
pure acks sent:	2		pure acks sent:	35791		
unique bytes sent:			unique bytes sent:	0		
actual data pkts:	72029		actual data pkts:	0		
actual data bytes:	104282744		actual data bytes:	0		
rexmt data pkts:	0		rexmt data pkts:	0		
rexmt data bytes:	0		rexmt data bytes:	0		
outoforder pkts:	0		outoforder pkts:	0		
pushed data pkts:	72029		pushed data pkts:	0		
SYN/FIN pkts sent:	1/1		SYN/FIN pkts sent:	1/1		
req 1323 ws/ts:	Y/Y		req 1323 ws/ts:	Y/Y		
adv wind scale:	0		adv wind scale:	4		
req sack:	Y O		req sack:	N O		
sacks sent:	-	Acres de la constantina della	sacks sent:	-	hh	
mss requested: max segm size:		bytes bytes	mss requested: max seqm size:		bytes bytes	
min segm size: avg segm size:		bytes bytes	min segm size: avg segm size:		bytes bytes	
max win adv:		bytes	max win adv:	750064		
min win adv:		bytes	min win adv:		bytes	
zero win adv:		times	zero win adv:		times	
avq win adv:	-	bytes	avg win adv:	-	bytes	
initial window:		bytes	initial window:		bytes	
initial window:		pkts	initial window:		pkts	
ttl stream length:			ttl stream length:		bytes	
missed data:		bytes	missed data:		bytes	
truncated data:	101833758		truncated data:		bytes	
truncated packets:	72029		truncated packets:		pkts	
data xmit time:	11.461		data xmit time:	0.000		
idletime max:	372.0	ms	idletime max:	246.8	ms	
throughput:	9097174	Bps	throughput:	0	Bps	
Dykstra, SC2002						

## TCP Connection Establishment

- Three-way handshake
  - SYN, SYN+ACK, ACK
- Use tcpdump, look for performance features
  - window sizes, window scale, timestamps,MSS, SackOK, Don't-Fragment (DF)

## Tcpdump of TCP Handshake

16:08:33.674226 wcisd.hpc.mil. 40874 > damp-nrl. 56117:

S 488615735:488615735(0) win 5840

<mss 1460,sackOK,timestamp 263520790 0,nop,wscale 0> (DF)

16:08:33.734045 damp-nrl.56117 > wcisd.hpc.mil.40874:

S 490305274:490305274(0) ack 488615736 win 5792

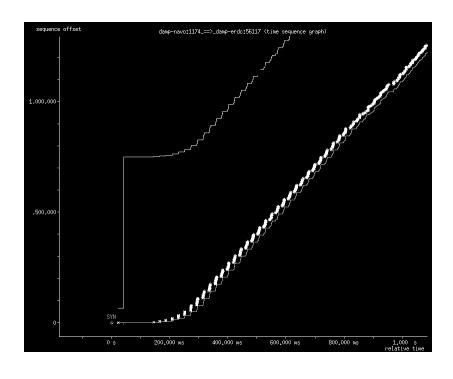
<mss 1460,sackOK,timestamp 364570771 263520790,nop,wscale 5> (DF)

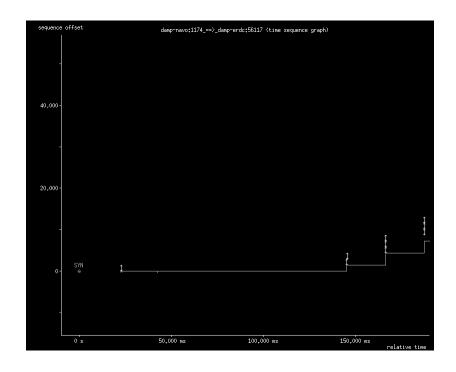
16:08:33.734103 wcisd.hpc.mil.40874 > damp-nrl.56117:

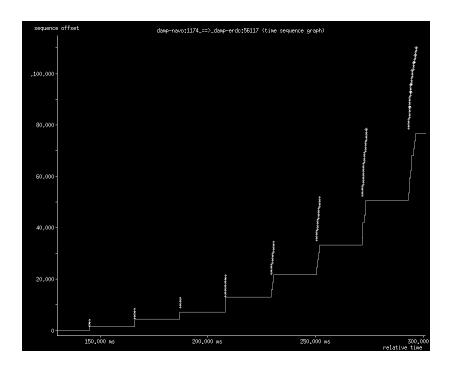
. ack 1 win 5840

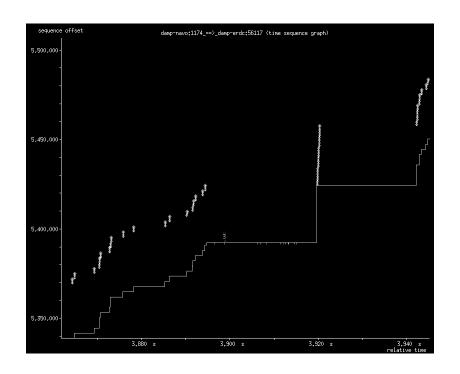
<nop,nop,timestamp 263520796 364570771> (DF)

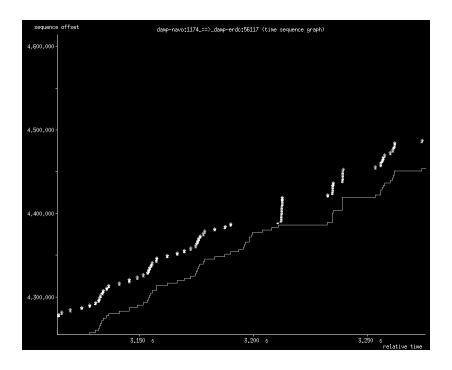
Dykstra, SC2002



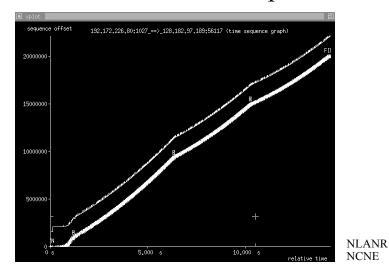








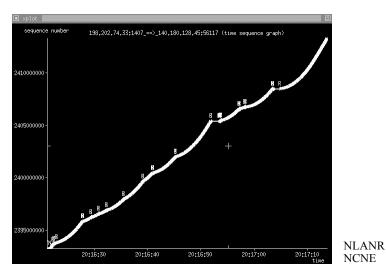
## Normal TCP Scallops



Dykstra, SC2002

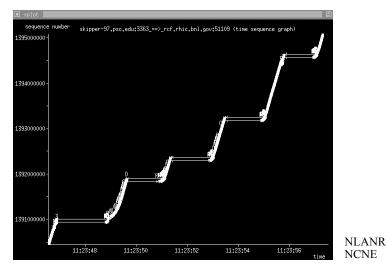
127

## A Little More Loss



Dykstra, SC2002

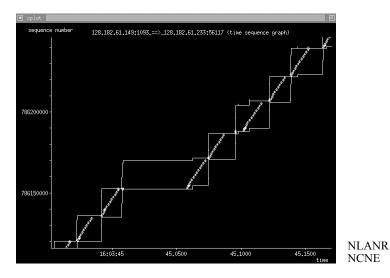
#### **Excessive Timeouts**



Dykstra, SC2002

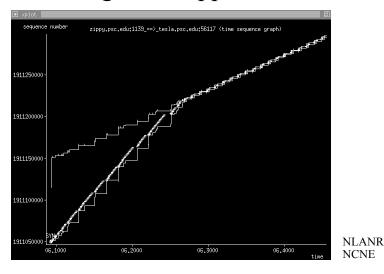
129

## Bad Window Behavior



Dykstra, SC2002

#### Receiving Host/App Too Slow



Dykstra, SC2002

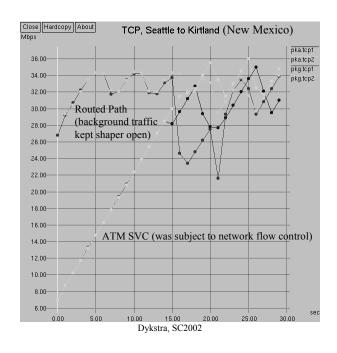
131

#### Traffic Management

Beware of network flow control

- ATM Forum TM 4.1
  - For UBR and ABR flows (Unspecified and Available Bit Rate)
  - Blocks cells from entering the WAN
  - "Opens up" over time if BW is available
- Can destroy the performance of short flows
- Similar TM mechanisms are being considered for MPLS!

Dykstra, SC2002



133

## nuttcp: One Second UDP Test

```
damp-kirt$ nuttcp -t -T1 -u -R30m damp-ssc2
nuttcp-t: v3.1.9: socket
nuttcp-t: buflen=8192, nstream=1, port=5001 udp -> damp-ssc2
nuttcp-t: time limit = 1 second
nuttcp-t: rate limit = 30.000 Mb/s
nuttcp-t: send window size = 65535, receive window size = 65535
nuttcp-t: 3.578 MB in 1.00 real seconds = 3662.08 KB/sec = 29.9997 Mb/s
nuttcp-t: 464 I/O calls, msec/call = 2.21, calls/sec = 463.76
nuttcp-t: 0.5user 0.4sys 0:01real 100% 0i+0d 0maxrss 0+0pf 0+0csw
nuttcp-r: v3.1.9: socket
nuttcp-r: buflen=8192, nstream=1, port=5001 udp
nuttcp-r: send window size = 65535, receive window size = 65535
nuttcp-r: 0.609 MB in 2.02 real seconds = 309.21 KB/sec = 2.5331 Mb/s
nuttcp-r: 82.98% data loss
nuttcp-r: 80 I/O calls, msec/call = 25.83, calls/sec = 39.64
nuttcp-r: 0.0user 0.0sys 0:02real 0% 0i+0d 0maxrss 0+0pf 0+0csw
                                Dykstra, SC2002
                                                                       134
```

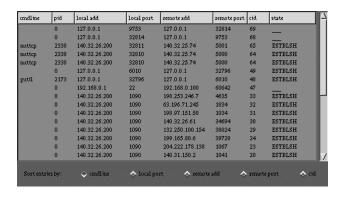
#### Web100

#### www.web100.org

- Set out to make 100 Mbps TCP common
- "TCP knows what's wrong with the network"
  - Mostly on the sender side
- Instruments the TCP stack for diagnostics
- Enhanced TCP MIB (IETF Draft)
- Linux 2.4 kernel patches + library and tools
- /proc/web100 file system
   e.g. /proc/web100/1010/{read,spec,spec-ascii,test,tune}

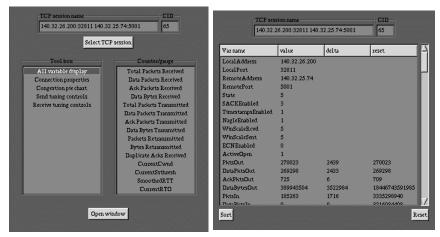
Dykstra, SC2002

#### Web100 – Connection Selection



Dykstra, SC2002

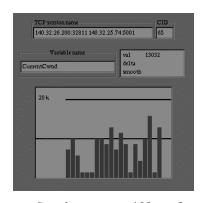
# Web100 - Tool/Variable Selection

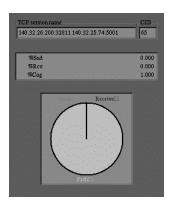


Dykstra, SC2002

137

## Web100 – Variable Display, Triage Chart





See also www.net100.org for more work based on Web100

Dykstra, SC2002

# ANL Network Tester <a href="http://miranda.ctd.anl.gov:7123/">http://miranda.ctd.anl.gov:7123/</a>



Dykstra, SC2002

139

#### **ANL Tester - Statistics**

Dykstra, SC2002

## Iperf with Web100, Clean Link

wcisd\$ iperf-web100 -e -w400k -p56117 -c damp-wcisd

Client connecting to damp-wcisd, TCP port 56117 TCP window size: 800 KByte (WARNING: requested 400 KByte)

- [ 3] local 192.168.26.200 port 33185 connected with 192.168.26.61 port 56117 [ID] Interval Transfer Bandwidth [ 3] 0.0-10.0 sec 113 MBytes 94.1 Mbits/sec

Web100 Analysis -

100 Mbps FastEthernet link found Good network cable(s) found Duplex mismatch condition NOT found Link configured for Full Duplex operation Information: This link is congested with traffic

Web100 reports the Round trip time = 14.0 msec; the Packet size = 1448 Bytes; and There were 1 packets retransmitted, 0 duplicate acks received, and 0 SACK blocks received This connection is network limited 99.99% of the time. Contact your local network administrator to report a network problem

Web100 reports the Tweakable Settings are: RFC-1323 Time Stamping: On RFC-1323 Window Scaling Option: On RFC-2018 Selective Acknowledgment (SACK): On

Dykstra, SC2002

#### Iperf with Web100, Lossy Link

wcisd\$ iperf-web100 -e -w400k -p56117 -c damp-ssc2

Client connecting to damp-ssc2, TCP port 56117

TCP window size: 800 KByte (WARNING: requested 400 KByte)

- [ 3] local 192.168.26.200 port 33198 connected with 192.168.25.74 port 56117
- [ ID] Interval Transfer Bandwidth
- [ 3] 0.0-10.2 sec 35.0 MBytes 28.9 Mbits/sec

---- Web100 Analysis -----

Unknown link type found Good network cable(s) found

Warning: Duplex mismatch condition exists: Host HD and Switch FD Information: link configured for Half Duplex operation

 $Web100 \ reports \ the \ Round \ trip \ time = 2.0 \ msec; \ the \ Packet \ size = 1448 \ Bytes; \ and \ There \ were \ 617 \ packets \ retransmitted, \ 4072 \ duplicate \ acks \ received, \ and \ 4370 \ SACK \ blocks \ received \ and \ blocks \ received \ blocks \ received$ The connection stalled 1 times due to packet loss The connection was idle for 0.21 seconds (2.06%) of the time This connection is network limited 99.99% of the time. Contact your local network administrator to report a network problem

Dykstra, SC2002

## Going Faster

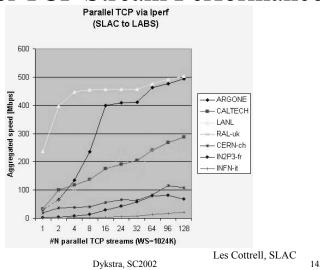
Cheating Today, Improving TCP
Tomorrow

#### Parallel TCP Streams

- PSockets (Parallel Sockets library)
  - http://citeseer.nj.nec.com/386275.html
- GridFTP enhanced parallel wu-ftpd
  - http://www.globus.org/datagrid/gridftp.html
- bbFTP parallel 'ftp' uses ssh or GSI
  - http://doc.in2p3.fr/bbftp/
- MulTCP a TCP that acts like N TCP's
   UCL

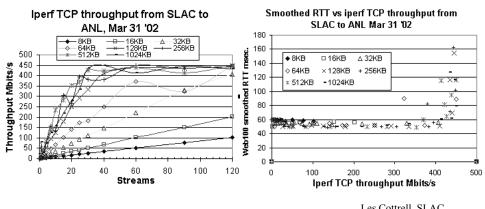
Dykstra, SC2002

## Parallel TCP Stream Performance



#### Parallel TCP Streams

#### Throughput and RTT by Window Size



Les Cottrell, SLAC

Dykstra, SC2002 146

#### Tsunami - Use UDP Instead

http://www.anml.iu.edu/

- UDP data stream, constant rate
- TCP control stream to request retransmissions
- Transferred 1 TB of data at ~1 Gbps over a 12000 km "light path" (Vancouver to Geneva), Sep 2002
- Was created because TCP over that path was getting only 10's to 100's of Mbps!

Dykstra, SC2002

147

#### The Future of TCP/IP

- Different retransmit/recovery schemes
   TCP Taho, Vegas, Peach, Westwood, ...
- Pacing removing burstiness by spreading the packets over a round trip time (BLUE)
- Rate-halving to recover ACK clocking more quickly after loss
- Autotuning buffer space usage

#### The Future of TCP/IP cont.

- Limited Transmit, RFC 3042 open window on duplicate ACKs (Proposed Standard Jan 2001)
- Explicit Congestion Notification (ECN)
   RFC 3168, Sep 2001
- Modifications to prevent "cheating"
- Kick-starting TCP after timeouts

Dykstra, SC2002

149

#### **Increased Initial Windows**

draft-ietf-tsvwg-initwin-04.txt

- Allows ~4KB initial window rather than one or two segments
  - min(4\*MSS, max(2\*MSS, 4380 bytes))
- Proposed Standard Aug 28, 2002.

#### **Appropriate Byte Counting**

draft-allman-tcp-abc-03.txt

- When an ACK is received, increase cwin based on the *number of new bytes ACK'd*
- Prevents receiver from "cheating" and making the sender open cwin too quickly
  - e.g. receiver ACKs every byte
- Increases by at most 2\*MSS bytes per ACK
  - To avoid bursts when one ACK covers a huge number of bytes

Dykstra, SC2002

151

## Quick-Start

draft-amit-quick-start-01.txt

- IP option in the TCP SYN specified desired initial sending rate
  - Routers on the path decrement a TTL counter and decrease initial sending rate if necessary
- If all routers participated, receiver tells the sender the initial rate in the SYN+ACK pkt
- The sender can set cwin based on the rtt of the SYN and SYN+ACK packets

#### Limited Slow-Start

#### www.icir.org/floyd/hstcp.html

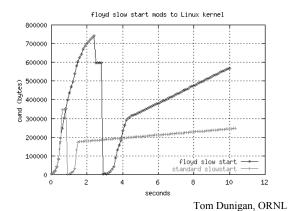
- In slow-start, the congestions window (cwin) doubles each round trip time
- For large cwins, this doubling can cause massive packet loss (and network load)
- Limited slow-start adds *max\_ssthresh* (proposed value of 100 MSS)
- Above *max\_ssthresh* cwin opens slower, never bursts more than 100 MSS

cwin += (0.5\*max ssthresh/cwin) \* MSS

Dykstra, SC2002

153

## Limit Slow-Start Example



Dykstra, SC2002

## HighSpeed TCP

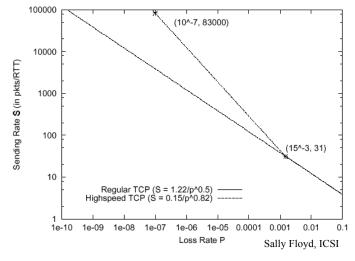
#### www.icir.org/floyd/hstcp.html

- Changes the AIMD parameters
- Identical to standard TCP for loss rates below 10<sup>-3</sup> for fairness (cwin <= 38)
- Allows cwin to reach 83000 segments for 10<sup>-7</sup> loss rates
  - Good for 10 Gbps over 100 msec rtt
  - Std TCP would be limited to ~440 Mbps

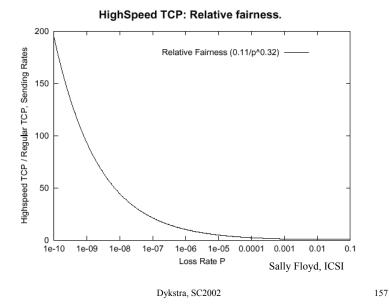
Dykstra, SC2002

155

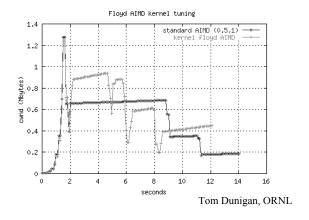
#### HighSpeed TCP: use a modified response function.



Dykstra, SC2002

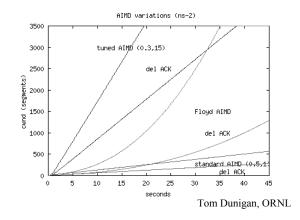


## Standard vs. HighSpeed TCP



Dykstra, SC2002

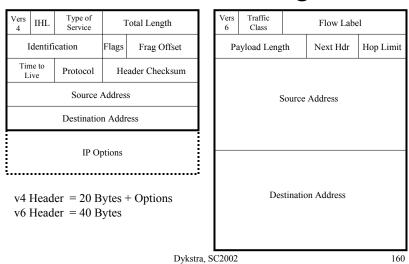
## **Cwin Opening Comparison**



Dykstra, SC2002

159

## IPv4 to IPv6 Changes



# Multi-Protocol Label Switching (MPLS)

- Adds switched (layer 2) paths to below IP
  - Useful for traffic engineering, VPN's, QoS control, high speed switching
- IP packets get wrapped in MPLS frames and "labeled"
- MPLS routers switch the packets along Label Switched Paths (LSP's)
- Being generalized for optical switching

Dykstra, SC2002

161

#### Review

- Network capacity vs. speed
- Importance of window and buffer sizes
- How TCP throughput depends on delay, loss, packet size
- How to use ping, traceroute, treno, etc.
- Looking deeper for problems
- TCP/IP keeps evolving

#### Recommended Resources

- · Richard W. Stevens' books
  - TCP/IP Illustrated, ISBN 0-201-63346-9
  - <a href="http://www.kohala.com/start/">http://www.kohala.com/start/</a>
- Host performance tuning details
  - <a href="http://www.psc.edu/networking/perf">http://www.psc.edu/networking/perf</a> tune.html
- CAIDA Internet Measurement Tool Taxonomy
  - http://www.caida.org/tools/

Dykstra, SC2002

163

#### Recommended Resources

- Iperf for TCP and UDP throughput testing
  - $-\ http://dast.nlanr.net/Projects/Iperf/$
- Testrig for TCP traces
  - http://ncne.nlanr.net/research/tcp/testrig/
- Web 100
  - http://www.web100.org/

Dykstra, SC2002



## Thank You!



Phillip Dykstra
WareOnEarth Communications Inc.
2109 Mergho Impasse
San Diego, CA 92110
phil@sd.wareonearth.com
619-574-7796